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EXAMINER	
STORM, DONALD L	

ART UNIT	PAPER NUMBER
2654	

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Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)
	10/017,000	ISO-SIPILA, JUHA
	Examiner	Art Unit
	Donald L. Storm	2654

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *14 December 2001 through March 18, 2002*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-20* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-20* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *14 December 2001* is/are: a)☐ accepted or b)☒ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date *3/18/02*.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Drawings

1. Figures 1A and 1B are objected to because they are not designated by a legend such as --PRIOR ART--. The legend is necessary in order to clarify what Applicant's invention is because only that which is old is illustrated. See MPEP § 608.02(g). The specification describes Figs. 1A and 1B as prior art on page 8, lines 21 and 22.

2. A permanent replacement sheet (a minimum being a black ink sketch suitable for publication) in compliance with 37 CFR 1.121(d) containing at least the corrected, substitute drawing for each figure being corrected is required in response to this Office action. Any amended, replacement drawing sheet should include all of the figures appearing on the immediate prior version of the sheet, even if only one figure is being amended. The figure or figure number of an amended drawing should not be labeled as "amended." If a drawing figure is to be canceled, the appropriate figure must be removed from the replacement sheet, and where necessary, the remaining figures must be renumbered and appropriate changes made to the brief description of the several views of the drawings for consistency. Additional replacement sheets may be necessary to show the renumbering of the remaining figures. The replacement sheet(s) should be labeled "Replacement Sheet" in the page header (as per 37 CFR 1.84(c)) so as not to obstruct any portion of the drawing figures. If the changes are not accepted by the examiner, the applicant will be notified and informed of any required corrective action in the next Office action. Corrected drawing sheets may no longer be held in abeyance. REPLACEMENT SHEETS LESS THAN THE MINIMUM DESCRIBED ABOVE WILL NOT BE CONSIDERED A *BONA FIDE* ATTEMPT TO PROVIDE A COMPLETE REPLY. See 37 C.F.R. § 1.121(d), § 1.81(d), § 1.85(a), and MPEP § 608.02 IV.

Claim Informalities

3. Claim 17, and by dependency claims 18-19, are objected to under 37 CFR 1.75(a) because the meaning of the phrase "the generated speech features" (page 20, line 1) needs clarification. Because feature vectors were previously recited as generated, but speech features were previously recited as extracted, it may be unclear as to what element this phrase refers. To further timely prosecution and evaluate prior art, the Examiner has interpreted this phase to refer to --the generated feature vectors--.

4. Claim 17, and by dependency claims 18-19, are objected to under 37 CFR 1.75(a) because the meaning of the phrase "the speech features" (page 20, line 5) needs clarification. Because feature vectors were previously recited as generated, but generated speech features were previously recited and speech features were previously recited as extracted, it may be unclear as to what element this phrase refers. To further timely prosecution and evaluate prior art, the Examiner has interpreted this phase to refer to --the extracted speech feature--.

5. Claim 17, and by dependency claims 18-19, are objected to under 37 CFR 1.75(a) because the meaning of the phrase "the noise-reduction feature vectors" (page 20, lines 5-6) needs clarification. Because spectral magnitude values were previously recited as the first signal for reducing noise, and the feature vectors were previously recited merely as having noise, but not as corresponding to noise reduction, it may be unclear as to what element this phrase refers. To further timely prosecution and evaluate prior art, the Examiner has interpreted this phase to refer to --the noise-reduced feature vectors--.

6. Claim 20 is objected to under 37 CFR 1.75(a) because ends with two periods. Each claim begins with a capital letter and ends with a period to avoid undue confusion in determining if the claim is complete. Appropriate correction is required. See MPEP § 608.01(m).

Claim Rejections - 35 USC § 102

7. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

Higgins

8. Claims 1, 3-10, 12, 14, 16-17, and 19 are rejected under 35 U.S.C. 102(e) as being anticipated by Higgins et al. [US Patent 6,266,633].

9. Regarding claim 14, Higgins [at Fig. 1] describes the distributed speech recognition system for processing a speech signal by describing the content and functionality of the recited limitations recognizable as a whole to one versed in the art as the following terminology:

a front-end unit responsive to the speech signals [see Fig. 1, items 9, 18, 23, 26, and their descriptions especially at column 4, line 64-column 5, line 1, of the preprocessor provided with input from the user speaking into microphone producing speech utterances];

it extracts speech features from the speech signals and [at column 5, lines 40-44, as the preprocessor extracts spectral magnitude features from windowed speech utterances];

it provides a first signal indicative of the extracted features [at column 5, lines 43-64, as the preprocessor provides a signal converted from the spectral data of spectral magnitudes of the windowed speech];

a back-end responsive to the first signal for recognizing words representative of the speech signals and providing a second signal indicative of the recognized words [see Fig. 1, items 10, 16, 18, 23, 26, and their descriptions especially at column 5, lines 9-30, of the microphone providing speech to the sampler, preprocessor, and voice verification system, that includes the verify processor that enables accurate word recognition of phrases in the utterance and produces a decision based on the processed speech consisting of the words];

the front-end has means to normalize [at column 4, lines 54-56, as the preprocessor provides normalization];

it normalizes the extracted speech features [at column 5, lines 52-53, as the preprocessor subtracts from the magnitude spectra the noise floor and sets negative results to zero as the (spectral subtraction) SS-processed magnitude];

the front-end has means for filtering [at column 5, lines 52-56, as the preprocessor applies the blind deconvolution filter];

it filters the normalized speech features [at column 5, lines 52-56, as the preprocessor applies the blind deconvolution filter by multiplying the SS-processed magnitude and rejects frequencies];

it filters in order to reduce noise in the speech signal [at column 4, lines 54-64, as the preprocessor provides noise suppression of a noisy voice signals of speech utterances].

10. Regarding claim 16, Higgins also describes:

filtering by a data-driven filter [at column 7, lines 39-67, as the BD filter has response inversely proportional to the channel frequency response obtained from the magnitudes at frequencies of the utterance].

11. Regarding claim 17, Higgins [at Fig.. 1] describes the speech recognition feature extractor for extracting speech features for a speech signal by describing the content and functionality of the recited limitations recognizable as a whole to one versed in the art as the following terminology:

a time-to-frequency domain transformer generates spectral magnitude values in a frequency domain of the speech signal [at column 6, lines 4-55, as an FFT and converter module obtains magnitude spectra at frequencies indicative of a speech utterance];

it provides a first signal indicative of them [at column 6, lines 59-60, as each magnitude is converted to a real number];

a feature generator responsive to the first signal generates a plurality of feature vectors [at column 8, lines 51-54, as the processing computes spectral magnitudes for 512 frequencies for 46 frames];

it provides a second signal indicative of the generated speech features (vectors) [at column 8, lines 44-54, as the processing computes spectral magnitudes for each frame at each frequency for the FFTed and converted speech utterance];

normalizing means [at column 4, lines 54-56, as the preprocessor provides normalization];

it is responsive to the second signal and normalizes the generated feature vectors and provides a third signal indicative of the normalized speech features [at column 5, lines 52-53, as

the preprocessor subtracts from the magnitude spectra the noise floor and sets negative results to zero as the (spectral subtraction) SS-processed magnitude];

a frequency filtering means [at column 5, lines 52-56, as the blind deconvolution filter];

it is responsive to the first signal [at column 5, lines 52-56, as the preprocessor applies the blind deconvolution filter by multiplying the processed magnitudes that resulted from subtracting the noise floor from the magnitude spectra];

it reduces noise [at column 4, lines 54-64, as the preprocessor provides noise suppression of a noisy voice signals of speech utterances];

the noise is in the normalized feature vectors [at column 7, lines 59-61, as the output of SS provides a noise-suppressed signal];

it provides the (extracted) speech features indicative of the noise-reduced feature vectors [at column 5, lines 60-64, as the preprocessor then converts the spectral data after BD of the (spectral subtraction) SS-processed magnitude spectra].

12. Claim 19 sets forth additional limitations similar to limitations set forth in claim 16. Higgins describes the additional limitations as indicated there.

13. Regarding claim 9, Higgins [at Fig.. 1] describes the distributed speech recognition front-end by describing the content and functionality of the recited limitations recognizable as a whole to one versed in the art as the following terminology:

a first means responsive to a speech signal [see Fig. 1, items 9, 18, 23, 26, and their descriptions especially at column 4, line 64-column 5, line 1, of the preprocessor provided with input from the user speaking into microphone producing speech utterances];

it extracts speech features from the speech signal [at column 5, lines 40-44, as the preprocessor extracts spectral magnitude features from windowed speech utterances];

it provides a first signal indicative of the extracted features [at column 5, lines 43-44, as the preprocessor utilizes spectral magnitudes of the windowed speech];

second means for normalizing [at column 4, lines 54-56, as the preprocessor provides normalization];

it is responsive to the first signal and normalizes the extracted speech features and provides a second signal indicative of the normalized speech features [at column 5, lines 52-53, as the preprocessor subtracts from the magnitude spectra the noise floor and sets negative results to zero as the (spectral subtraction) SS-processed magnitude];

third means for filtering [at column 5, lines 52-56, as the preprocessor applies the blind deconvolution filter];

it is responsive to the second signal and filters the normalized speech features [at column 5, lines 52-56, as the preprocessor applies the blind deconvolution filter by multiplying the SS-processed magnitude and rejects frequencies];

it filters in the frequency domain [at column 7, line 65-column 8, line 24, as the BD filter operations repeat for each value of analysis frequencies];

it filters in order to reduce noise [at column 4, lines 54-64, as the preprocessor provides noise suppression of a noisy voice signals of speech utterances];

the noise is in the second signal [at column 7, lines 59-61, as the output of SS provides a noise-suppressed signal];

it provides a third signal indicative of the filtered speech features [at column 5, lines 60-64, as the preprocessor then provides a signal conversion of the spectral data after BD];

means for conveying the third signal to a back-end [at column 5, lines 60-64, as the preprocessor then converts the spectral data and provides a signal for processing by the verifying system];

the back-end is speech recognition for the back-end to recognize words representative of the speech signal from the third signal [see Fig. 1, items 10, 16, 26, and their descriptions especially at column 5, lines 9-30, of the speech through the preprocessor to the voice verification system, that includes the verify processor that enables accurate word recognition of phrases in the utterance by comparing the processed speech];

back-end is distributed speech recognition [see Fig. 1, items 10, 16, 23, 26, and their descriptions especially at column 5, lines 9-30, of the voice verification system, that includes the verify processor that enables accurate word recognition, follows the sampler and preprocessor].

14. Claim 10 sets forth additional limitations similar to limitations set forth in claim 16. Higgins describes the additional limitations as indicated there.

15. Regarding claim 12, Higgins also describes:

a time domain, preprocessing device to convert the speech signal to a digital signal [at column 4, lines 64-67, as an analog to digital converter sampling at a rate per second (Hz) to provide speech utterances as a digitized voice signal];

a time-to-frequency conversion device to provide a set of magnitude spectrum values for the digital signal [at column 6, lines 4-55, as an FFT and converter module to obtain magnitude spectra for each windowed frame of data samples];

an assembly device to assemble the set of magnitude spectrum values into the speech features [at column 8, lines 51-54, as the processing computes spectral magnitudes for 512 frequencies for 46 frames].

16. Regarding claim 1, Higgins [at Fig.. 1] describes the method for speech processing in a distributed-speech recognition system having a front-end and a back-end for recognizing words from speech signals by describing the content and functionality of the recited limitations recognizable as a whole to one versed in the art as the following terminology:

extracting speech features from the speech signals [at column 5, lines 40-44, as extract spectral magnitude features from windowed speech utterances];

the speech features contain a speech to noise ratio [at column 7, lines 20-40, as the magnitudes spectra evince background noise, a noise floor, and magnitudes that exceed the noise floor];

normalizing the speech features [at column 5, lines 52-53, as the subtract from the magnitude spectra the noise floor and sets negative results to zero as the (spectral subtraction) SS-processed magnitude];

filtering the normalized speech features [at column 5, lines 52-56, as apply the blind deconvolution filter by multiplying the SS-processed magnitude and rejects frequencies];

the filtering is in the frequency domain [at column 7, line 65-column 8, line 24, as the BD filter operations repeat for each value of analysis frequencies];

conveying the speech features forma the front end to the back-end [at column 5, lines 60-64, as the convert the spectral data and provides a signal for processing by the verifying system].

17. Claim 3 sets forth additional limitations similar to limitations set forth in claim 16. Higgins describes the additional limitations as indicated there.

18. Regarding claim 4, Higgins also describes:

converting the speech signals for a time domain to a frequency domain prior to extracting the speech features [at column 6, lines 4-55, as FFT and convert module to obtain magnitude spectra for speech data samples and computes spectral magnitudes for frequencies for each frame].

19. Regarding claim 5, Higgins also describes:

converting the speech signals to digital signals prior to converting from time to frequency [at column 4, lines 64-67 and column 5, line 40-44, as convert analog speech utterances to digital to provide speech as a digitized voice signal for input to a preprocessor to extract the spectral magnitudes].

20. Regarding claim 6, Higgins also describes:

it is carried out by a Fast Fourier Transform to compute a magnitude spectrum and provide a plurality of magnitude spectrum values [at column 6, lines 4-55, as an FFT and converter module to obtain magnitude spectra for each windowed frame of data samples];

it is carried out to provide a plurality of magnitude spectrum values [at column 8, lines 51-54, as the processing computes spectral magnitudes for 512 frequencies for 46 frames].

21. Regarding claim 7, Higgins also describes:

non-linearly modifying the magnitude spectrum and generating a plurality of logarithmically warped magnitude spectrum values [at column 7, line 45, as log m_{ft}].

22. Regarding claim 8, Higgins also describes:

assembling the logarithmically warped spectrum values [at column 7, line 45, as Σ log m_{ft} to determine C_f];

the assembling produces a set of features parameters representative of the speech features [at column 9, lines 32-43, as subtract from each m_{ft} the noise floor and set negative results to zero and then perform blind deconvolution with frequency response based on C_f].

Claim Rejections - 35 USC § 103

23. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

Higgins and Hermansky

24. Claims 2, 11, 15, and 18 are rejected under 35 U.S.C. 103(a) as being unpatentable over Higgins et al. [US Patent 6,266,633] in view of Hermansky, Herman, "RASTA Processing of Speech," IEEE Trans. Speech and Audio Proc., vol. 2, no. 4, October 1994, pp. 578-589, already of record.

25. Regarding claim 2, Higgins describes the included claim elements as indicated elsewhere in this Office action. Higgins [at column 10, lines 10-14] also points out that filtering may include conventional smoothing of the magnitude spectra. However, Higgins does not explicitly mention that smoothing is one function of a low pass filter.

Like Higgins, Hermansky [at § III] describes a system that removes noise from input speech by filtering in the spectrum domain to improve automatic speech recognition. Hermansky also includes smoothing of the signal, and describes:

filtering by a low-pass filter [at page 579, column 2, as low-pass filtering helps to smooth spectral changes].

Both Higgins and Hermansky [at § III] filter the time trajectory in the transformed spectral domain and the filter used by Higgins [at column 10, lines 11-14] and Hermansky [at § III] provides smoothing. As indicated, Hermansky shows that low-pass filtering for smoothing was known to artisans at the time of invention. Hermansky also points out that smoothing has the advantage of suppressing high frequency artifacts. To the extent that Higgins's filter that provides the smoothing is not necessarily a low-pass filter, it would have been obvious to one of ordinary skill in the art of noise suppression at the time of invention to include the concepts described by Hermansky at least filtering by a low-pass filter to provide Higgins's smoothing because low-pass filtering was know by artisans to reduce artifacts in the processed signal to help Higgins determine the appropriate noise magnitude.

26. Regarding claim 11, Higgins describes the included claim elements as indicated elsewhere in this Office action. The claim sets forth additional limitations similar to limitations set forth in

claim 2. Higgins and Hermansky describe and make obvious the additional limitations as indicated there.

27. Regarding claim 15, Higgins describes the included claim elements as indicated elsewhere in this Office action. The claim sets forth additional limitations similar to limitations set forth in claim 2. Higgins and Hermansky describe and make obvious the additional limitations as indicated there.

28. Regarding claim 18, Higgins describes the included claim elements as indicated elsewhere in this Office action. The claim sets forth additional limitations similar to limitations set forth in claim 2. Higgins and Hermansky describe and make obvious the additional limitations as indicated there.

Higgins and ETSI ES 201 108 V1.1.2

29. Claims 13 and 20 are rejected under 35 U.S.C. 103(a) as being unpatentable over Higgins et al. [US Patent 6,266,633] in view of ETSI ES 201 108 V1.1.2, already of record.

30. Regarding claim 13, Higgins describes the included claim elements as indicated elsewhere in this Office action. Higgins [at column 8, lines 41-67] describes an exemplary embodiment that discusses reduction in the sampling rate prior to passing the spectral data after BD to the speech recognition process. Higgins discussion is for the FFT process, and Higgins describes:

the third signal has a sampling rate [at column 8, lines 43-column 9, line 37, as the deconvolved sample sequence after blind deconvolution is performed on spectral magnitudes with sampling frequencies of 8000/1024].

However, Higgins does not explicitly describe reduction of sampling rate of the signal following conversion of the spectral data after blind deconvolution.

ETSI ES 201 108 V1.1.2 [at title] standardizes the passage of speech recognition feature data from front-end processing to back-end speech recognition, and ETSI ES 201 108 V1.1.2 describes:

means to reduce the sampling rate before conveying a signal that has the sampling rate to a distributed signal recognition back-end [at § 1, as the algorithm to provide a lower data rate bitstream for transmission from the front-end feature extraction to "back-end" speech recognition].

As indicated, ETSI ES 201 108 V1.1.2 shows that sampling rate reduction before conveying a signal to a distributed speech recognition back-end was known to artisans at the time of invention. Since ETSI ES 201 108 V1.1.2 [at § 1] also points out that bit rate reduction has the advantage of providing a lower data transmission rate, it would have been obvious to one of ordinary skill in the art of data transmission at the time of invention to include the concepts described by ETSI ES 201 108 V1.1.2 at least reduction of sampling rate to convey Higgins's speech features for recognition because that would allow the features to be conveyed from the front-end to the back-end speech recognition using a lower data rate channel.

31. Regarding claim 20, Higgins [at Fig.. 1] describes the communications device by describing the content and functionality of the recited limitations recognizable as a whole to one versed in the art as the following terminology:

voice input unit to allow a user to input speech signals to the device [see Fig. 1, especially items 9, 18, and their descriptions at column 4, lines 64-65 of user speaking into microphone producing speech utterances];

means for providing speech data to an apparatus that includes a speech recognition back-end capable of recognizing speech based on the speech data [see Fig. 1, items 10, 16, 18, 23, 26, and their descriptions especially at column 5, lines 9-30, of the microphone providing speech to the sampler, preprocessor, and voice verification system, that includes the verify processor that enables accurate word recognition of phrases in the utterance];

the apparatus includes distributed speech recognition [see Fig. 1, items 10, 16, 23, 26, and their descriptions especially at column 5, lines 9-30, of the sampler and preprocessor followed by the voice verification system, that includes the verify processor that enables accurate word recognition];

a front-end unit responsive to the (input) speech signals [see Fig. 1, items 9, 18, 23, 26, and their descriptions especially at column 4, line 64-column 5, line 1, of the preprocessor provided with input from the user speaking into microphone producing speech utterances];

it extracts speech features from the speech signals and [at column 5, lines 40-44, as the preprocessor extracts spectral magnitude features from windowed speech utterances];

it provides a first signal indicative of the extracted features [at column 5, lines 43-44, as the preprocessor utilizes spectral magnitudes of the windowed speech];

it includes means for normalizing [at column 4, lines 54-56, as the preprocessor provides normalization];

it is responsive to the first signal and normalizes the extracted speech features and provides a second signal indicative of the normalized speech features [at column 5, lines 52-53, as the

preprocessor subtracts from the magnitude spectra the noise floor and sets negative results to zero as the (spectral subtraction) SS-processed magnitude];

it includes means for filtering [at column 5, lines 52-56, as the preprocessor applies the blind deconvolution filter];

it is responsive to the second signal and filters the normalized speech features [at column 5, lines 52-56, as the preprocessor applies the blind deconvolution filter by multiplying the SS-processed magnitude and rejects frequencies];

it filters in order to reduce noise in the speech signal [at column 4, lines 54-64, as the preprocessor provides noise suppression of a noisy voice signals of speech utterances];

it includes the filtered speech features in the (provided-to-the-apparatus) speech data [at column 5, lines 60-64, as the preprocessor then converts the spectral data and provides a signal (for processing by the verifying system)].

Higgins does not explicitly describe that the recognition back-end is provided by a external apparatus including distributed speech recognition.

ETSI ES 201 108 V1.1.2 [at title] standardizes the passage of speech recognition feature data from front-end processing to back-end speech recognition, and ETSI ES 201 108 V1.1.2 describes:

an external apparatus including a distributed speech recognition back-end [at Introduction, as a recognizer of a speech recognition system remote from a front-end terminal].

As indicated, ETSI ES 201 108 V1.1.2 shows that an external apparatus including a distributed speech recognition back-end was known to artisans at the time of invention. Since ETSI ES 201 108 V1.1.2 [at Introduction] also points out that DSR has the advantage of overcoming the problem of degradation of speech transmitted over mobile channels, it would have

been obvious to one of ordinary skill in the art of DSR at the time of invention to include the concepts described by ETSI ES 201 108 V1.1.2 at least an external apparatus including a distributed speech recognition back-end for Higgins' distributed speech recognition system because Higgins could provide DSR over mobile channels without degradation of speech by the channel during transmission.

Conclusion

32. The following references here made of record are considered pertinent to applicant's disclosure:

Zingher [US Patent 6,092,039] describes transmission of normalized, mel cepstrum, compressed speech feature vectors, filtered, from a cellular telephone to speech recognition on a server.

Mauuary et al. [US Patent 6,157,909] describes transmission of normalized, filtered, mel cepstrum, compressed speech feature vectors from a cellular telephone to speech recognition on a server.

Kingsbury et al. [US Patent 6,308,155] describes normalized speech spectral vectors, filtered by data-driven low-pass and high-pass filters to reduce noise.

Garudadri et al. [US Patent Application Publication 2003/0004720] describes voice recognition on a back-end of mel cepstral speech feature vectors received over a wireless channel

33. Any response to this action should be mailed to:

Mail Stop Amendment

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

or faxed to:

(703) 872-9306, (for formal communications intended for entry)

Or:

(703) 872-9306, (for informal or draft communications, and please label "PROPOSED" or "DRAFT")

Patent Correspondence delivered by hand or delivery services, other than the USPS, should be addressed as follows and brought to U.S. Patent and Trademark Office, 220 20th Street S., Customer Window, **Mail Stop Amendment**, Crystal Plaza Two, Lobby, Room 1B03, Arlington, VA, 22202

34. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Donald L. Storm, of Art Unit 2654, whose telephone number is (703) 305-3941. The examiner can normally be reached on weekdays between 8:00 AM and 4:30 PM Eastern Time. If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on (703) 305-9645.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Inquiries regarding the status of submissions relating to an application or questions on the Private PAIR system should be directed to the Electronic Business Center (EBC) at 866-217-9197 (toll-free) or 703-305-3028 between the hours of 6 a.m. and midnight Monday through Friday EST, or by e-mail at: ebc@uspto.gov. For general information about the PAIR system, see http://pair-direct.uspto.gov.

Donald L. Storm
Donald L. Storm
Patent Examiner
Art Unit 2654

December 9, 2004